DEFENSE INFORMATION SYSTEMS AGENCY
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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) 9 June 2008

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Extension of the Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.2(3) Service Release (SR) 3, with Internetwork Operating System (IOS) Software Release 12.4(9) T1

References: (a) DoD Directive 4630.5, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01D, "Interoperability and Supportability of Information Technology and National Security Systems," 8 March 2006

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification. Additional references are provided in the enclosure.

2. The CCM Version 4.2(3) SR3 with IOS Software Release 12.4(9) T1 is hereinafter referred to as the system under test (SUT). The SUT meets all of the critical interoperability requirements and is certified for joint use within the Defense Switched Network (DSN) for the following switch types: Private Branch Exchange (PBX) 1 and PBX 2. The SUT meets the VoIP critical interoperability requirements with any certified Assured Services Voice Application Local Area Network (ASVALAN) on the DSN Approved Products List (APL). The identified test discrepancies shown in the SUT Interoperability Summary that remained open after software patches were applied and regression testing was completed have an overall minor operational impact. No other configurations, features, or functions, except those cited within this report, are certified by the JITC, or authorized by the Program Management Office for use within the DSN. This certification expires upon changes that could affect interoperability, but no later than three years from the date of the original memorandum (4 March 2008).

3. This is a certification based on a desktop review of the SUT. The original certification was granted based on interoperability testing conducted by JITC, review of vendor's Letters of Compliance (LoC), and review of patches applied to the SUT. Interoperability testing of the CCM Version 4.2(3) with IOS Software Release 12.4(9) T1 was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 30 October 2006 through 6 January 2007 and documented in reference (c). Review of vendor's LoC was completed on 6 February 2007. Regression testing of CCM Version 4.2(3) SR3 with IOS Software Release 12.4(9) T1 was conducted from 31 October through 3 November 2007. Regression testing of the 7906G, 7942G, 7945G, 7962G, 7965G, and 7975G internet protocol telephones was conducted from 27 through 31 December 2007. A desktop review was requested

to include the Cisco Unified IP Phone Expansion Module 7914. The desktop review was completed on 22 May 2008. JITC analysis determined the Cisco Unified IP Phone Expansion Module 7914 is certified for joint use within the DSN.

4. The SUT certified hardware and software components are listed in table 1. The interoperability test summary of the SUT is indicated in table 2. The PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in table 3. This interoperability test status is based on the PBX 1's ability to meet:

a. DSN services for Network and Applications specified in reference (d).

b. PBX 1 interface and signaling requirements for trunks/lines specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

c. PBX 1 CRs/FRs specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

d. The overall system interoperability performance derived from test procedures listed in reference (f).

e. Internet Protocol version 6 requirements specified in reference (e), paragraph 1.7, table 1-4, verified through vendor submission of LoC.

Table 1. SUT Hardware and Software Components

CCM Version 4.2(3) SR3, with IOS Software Release 12.4(9) T1			
Component (See note 1.)	**Release**	**Sub-component**	**Function**
CallManagers **MCS7835H, MCS7835I1, MCS7835H2, MCS7825H2, MCS7825H3, MCS7835H1**, MCS7825H, MCS 7835I, MCS7845H, MCS7845I, MCS7825-H1, MCS7825I1, MCS7845H1, MCS7845I1	CCM 4.2(3) SR3	Not Applicable	Processing/Signaling
Cisco 3745/3725 Multiservice Access Router (Gateway) (See note 2.)	IOS 12.4(9) T1	**NM HD 2V**	TDM Interface NM HD Voice, 2-slot IP communications voice/fax
		NM HD 2VE	TDM Interface NM HD Voice, 2-slot IP communications enhanced voice/fax
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, Direct Inward Dial
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
		VWIC 1MFT T1	Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1
		VWIC 2MFT T1	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
		VWIC 2MFT T1 DI	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1, Drop and Insert
Cisco 3845/3825 Integrated Services Router (Gateway)	IOS 12.4(9) T1	**NM HDV2**	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers E1 (See note 3.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 4.)
		EVM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/ E1 (See note 3.)

Table 1. SUT Hardware and Software Components (continued)

CCM Version 4.2(3) SR3 with IOS Software Release 12.4(9) T1 (continued)			
Component (See note 1.)	**Release**	**Sub-component**	**Function**
Cisco 2851 Integrated Services Router (Gateway)	IOS 12.4(9) T1	NM HDV2	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		EVM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 4.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 4.)
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
CP-7940G and CP-7960G	Load: P00308000700	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7970G and CP-7971G-GE	Load: SCCP70.8-3-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7911G and CP-7906G	Load:SCCP11.8-3-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	Load: SCCP41.8-3-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7942G and CP-7962G	Load: SCCP42.8-3-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7945G and CP-7965G	Load: SCCP45.8-3-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7975G	Load: SCCP75.8-3-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
7914	Load: S000105000300	Not Applicable	Expansion module

LEGEND:

10/100BaseT - 10/100 Mbps (Baseband Operation, Twisted Pair) Ethernet
CCM - Cisco CallManager
CP - Cisco Phone
DI - Drop and Insert
DID - Direct Inward Dialing
DSN - Defense Switched Network
E1 - European Basic Multiplex Rate (2.048 Mbps)
EM - Expansion Module
EVM - Extension Voice Module
Fax - facsimile
FXS - Foreign Exchange Station
G - 10/100BaseT Ethernet (A Cisco part designator on their IP phone.)
GE - Gigabit Ethernet (A Cisco part designator on their IP phone.)
GSCR - Generic Switching Center Requirements
HD - High Density
HDA - High Density Analog
IOS - Internetwork Operating System
IP - Internet Protocol
IPv6 - Internet Protocol version 6
JITC - Joint Interoperability Test Command
Mbps - Megabits per second
MCS - Media Convergence Server
MFT - Multiflex Trunk
NM - Network Module
PBX 1 - Private Branch Exchange 1
PMO - Program Management Office
RJ - Registered Jack
SCCP - Skinny Client Control Protocol
SR - Service Release
SUT - System Under Test
T1 - Digital Transmission Link Level 1 (1.544 Mbps)
TDM - Time Division Multiplexing
V - Voice
VE - Voice/Fax Enhanced
VIC - Voice Interface Card
VWIC - Voice WAN Interface Card
WAN - Wide Area Network

NOTES:

1 Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same IOS software and hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.
2 All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745 and 3725. The 3745 and 3725 do not meet the critical IPv6 capability requirement in accordance with the GSCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:
   a. The component must already be JITC certified and currently fielded within the DSN.
   b. There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers.
3 These components support both T1 and E1; however, the E1 interface was not tested by JITC and is not approved for use within the DSN by the PMO. Since E1 interfaces are not required for a PBX 1, the risk of not testing is minor.
4 The EM HDA 8FXS expansion module requires the EVM HD module. Up to two EM HDA 8FXS expansion modules are supported for each EVM HD.

Table 2. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	Although the SUT supports T1 CAS, due to critical interoperability discrepancies discovered during testing, it is not certified. Wink start recognition is not within the required tolerance.[1] An off-hook seizure below the minimum limit is treated as valid.[2] A call fails to complete after trunk preemption.[3] Calls that are attempted over a trunk that is broken or in a remote busy-out condition do not receive ICA.[4] This interface is not certified by JITC and is not authorized for use within the DSN by the PMO. There is no overall operational impact because T1 CAS is not a required interface for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface was not tested by JITC and is not authorized for use within the DSN by the PMO. Since E1 interfaces are not required for a PBX 1, the risk of not testing is minor.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs with the following minor exception: Calls that are attempted over a trunk that is broken or in a remote busy-out condition do not receive an ICA.[4] The operational impact is minor.
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	This interface was not tested by JITC and is not authorized for use within the DSN by the PMO. Since E1 interfaces are not required for a PBX 1, the risk of not testing is minor.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs with a minor configuration change[5] and the following minor exceptions: The SUT does not support an MLPP global diversion number.[6] The BNEA is not provided.[7] The operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported. This is not a required interface for a PBX 1.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported. This is not a required interface for a PBX 1.
VoIP	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT does not support an MLPP global diversion number.[6] The BNEA is not provided.[7] The operational impact is minor.
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	No	Certified	Met all CRs and FRs with the following minor exceptions: Full compliance of DSN Common Call Features was not met.[8, 9, 10, 11, 12, 13] The operational impact is minor.
Attendant	No	Not Tested	This feature is not supported. This is not a required feature for a PBX 1.
Public Safety	No	Certified	All public safety features are conditional. The SUT met all CRs and FRs for E911. The SUT does not support the other public safety features. These features are not required for a PBX 1.[14]
Preset Conferencing	No	Not Tested	This feature is not supported. This is not a required feature for a PBX 1.
Nailed-up Connections	No	Not Tested	This feature is not supported. This is not a required feature for a PBX 1.
PAT	No	Not Tested	This feature is not supported. This is not a required feature for a PBX 1.
DSN Hotline Services	No	Not Tested	This feature is not supported. This is not a required feature for a PBX 1.
Network Management	No	Not Tested	This feature is not supported. This is not a required feature for a PBX 1.
ISDN Services (EKTS)	No	Not Tested	This feature is not supported. This is not a required feature for a PBX 1.
Synchronization	Yes	Certified	Met all CRs and FRs.[15]
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 16.	See note 16.
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASVALAN posted on the DSN APL. See notes 17 and 18.

Table 2. SUT Interoperability Test Summary (continued)

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	Although the SUT supports T1 CAS, due to critical interoperability discrepancies discovered during testing, it is not certified. Wink start recognition is not within the required tolerance.[1] An off-hook seizure below the minimum limit is treated as valid.[2] A call fails to complete after trunk preemption.[3] There is no overall operational impact because T1 CAS is not a required interface for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface was not tested by JITC and is not authorized for use within the DSN by the PMO. Since E1 interfaces are not required for a PBX 1, the risk of not testing is minor.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Tested	This interface was not tested by JITC and is not authorized for use within the DSN by the PMO. Since E1 interfaces are not required for a PBX 1, the risk of not testing is minor.
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified[19]	Met all critical CRs and FRs.

LEGEND:

ANSI	- American National Standards Institute	JITC	- Joint Interoperability Test Command
APL	- Approved Products List	LSSGR	- Local Access and Transport Area (LATA) Switching Systems Generic Requirements
ASVALAN	- Assured Services Voice Application Local Area Network	Mbps	- Megabits per second
BNEA	- Busy Not Equipped Announcement	MFR1	- Multi-Frequency Recommendation 1
BRI	- Basic Rate Interface	MLPP	- Multi-Level Precedence and Preemption
CAS	- Channel Associated Signaling	ms	- milliseconds
CRs	- Capability Requirements	NI 1/2	- National ISDN Standard 1 or 2
DISA	- Defense Information Systems Agency	PAT	- Precedence Access Threshold
DP	- Dial Pulse	PBX 1	- Private Branch Exchange 1
DRSN	- Defense Red Switch Network	PM	- Program Manager
DSN	- Defense Switched Network	PNT	- Preempt Notification Tone
DSS1	- Digital Subscriber Signaling 1	PRI	- Primary Rate Interface
DTMF	- Dual Tone Multi-Frequency	PSTN	- Public Switched Telephone Network
E1	- European Basic Multiplex Rate (2.048 Mbps)	Q.931	- Signaling Standard for ISDN
E911	- Basic Emergency Service 911	Q.955.3	- ISDN Signaling standard for E1 MLPP
EKTS	- Electronic Key Telephone System	SS7	- Signaling System 7
FRs	- Feature Requirements	SUT	- System Under Test
GR	- Generic Requirement	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
GR-506-CORE	- LSSGR: Signaling for Analog Interfaces	T1.607	- ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
GSCR	- Generic Switching Center Requirements	T1.619a	- SS7 and ISDN MLPP Signaling Standard for T1
ICA	- Isolated Code Announcement	TDM	- Time Division Multiplexing
IPv4	- Internet Protocol version 4	TPC	- Twisted Pair Copper
IPv6	- Internet Protocol version 6	VoIP	- Voice over Internet Protocol
ISDN	- Integrated Services Digital Network		
ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector		

Table 2. SUT Interoperability Test Summary (continued)

NOTES:

1 T1 CAS wink start recognition is not within the required tolerance of 100 ms to 350 ms. The SUT will only recognize a wink from 140 ms to 280 ms. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required for a PBX 1.

2 The SUT will treat any off-hook condition (ABCD Channel Associated Signaling bits high) of 12 ms or greater as a valid off-hook seizure and respond with a wink. In accordance with the requirements, signals that are less than 60 ms should be considered invalid. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required for a PBX 1.

3 During a trunk preemption test over the T1 CAS from the far-end to the SUT, after the preemption occurred the call would fail to complete and no treatment was provided to the call originator. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required for a PBX 1.

4 Calls that are attempted over a trunk that is broken or in a remote busy-out condition do not receive an Isolated Code Announcement (ICA). The operational impact is minor because they are treated with a Blocked Precedence Announcement (BPA) and since a PBX 1 cannot support special command and control users, the operational impact is mitigated.

5 To meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal, a configuration change was required on the analog gateways. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.

6 The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number. The operational impact is minor because they can configure the diversion settings for all of the stations provisioned on the switch from a single location.

7 When a station classmarked by the SUT as non-preemptable is active with a call and a higher precedence call attempts to directly preempt it, the BNEA is not provided. The operational impact is minor because the call is forwarded to the MLPP alternate directory number that is specified in the station's configuration.

8 Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP stations only. The following common call features are not supported by the SUT and therefore are not covered in this certification: Call-Waiting, Selective Call Rejection, and Denied Originating Service. These features are not required for a PBX 1.

9 All of the features on the IP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features.

10 Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP stations. This provides the ability for a user to receive additional calls while active with another call. There is no operational impact.

11 A short "ping" ring is not provided when calls are forwarded. There is a minor operational impact.

12 A conference disconnect tone is not provided when a three-way conference originator is preempted. This only occurs when an analog station originates the first call. The operational impact is minor because the preempted user receives Preempt Notification Tone (PNT) and the other members remain connected.

13 When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. The GSCR requires this only for Precedence above ROUTINE calls. There is no operational impact.

14 The SUT only supports E911 public safety features. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because public safety features are not required for a PBX 1.

15 To meet this requirement, a direct T1 interface must be connected between multiple gateways to synchronize timing of all TDM-based interfaces between gateways.

16 Security is tested by DISA-led Information Assurance test teams and published in a separate report.

17 An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria by 30 June 2008:

a. Conformant with IPv6 standards profile contained in the Department of Defense Information Technology Standards Registry (DISR).
b. Maintaining interoperability in heterogeneous environments and with IPv4.
c. Commitment to upgrade as the IPv6 standard evolves.
d. Availability of contractor/vendor IPv6 technical support.

18 All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745 and 3725. The 3745 and 3725 do not meet the critical IPv6 capability requirement in accordance with the GSCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:

a. The component must already be JITC certified and currently fielded within the DSN.
b. There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers.

19 Interoperability certification of the SUT does not constitute DRSN PM approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.

Table 3. PBX 1 Requirements

Interface	Critical	Requirements Required or Conditional		References
DSN Trunk Interfaces				
		Trunking	• Framing (R)	• GSCR Section 7
			• Line Code (R)	• GSCR Section 7
T1 CAS (MFR1, DTMF, DP)	No		• Signaling (R)	• GSCR Section 5
			• Alarm and Restoral Requirements (R)	• GSCR Section 7.1.4
			• Alarm and Restoral Requirements (C)	• GSCR Section 7.2.2
			• WWNDP (R)	• GSCR Section 4.5.1
			• Outpulsing digit formats	• GSCR Section 4.5.2
			• Routing (C)	• GSCR Section 4.2
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		• Trunk Groups (C)	• GSCR Section 2.5.5 & 2.5.6
			• Call Processing (R)	• GSCR Section 4 applicable paragraphs
			• CAS to CCS trunk interworking (C)	• GSCR Section 3.10
			• PCM-24/PCM-30 Interoperation (C)	• GSCR Section 7.3
			• Direct Inward Dialing (C)	• GSCR Section 2.3.2
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Voice	• MOS (R)	• CJCSI 6215.01B
			• MLPP (R)	• GSCR Section 3 applicable paragraphs
			• Secure calls (R)	• CJCSI 6215.01B
		Facsimile	• Analog: TIA/EIA-465-A (R)	• DISR
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Data	• Modem (VBD) (R)	• CJCSI 6215.01B
			• 56 kbps switched data (C: PRI only)	• GSCR Section 3.10
			• 64 kbps switched data (C: PRI only)	• GSCR Section 3.10
			• NX56 synchronous BER (C: PRI only)	• GSCR Section 3.10
			• NX64 synchronous BER (C: PRI only)	• GSCR Section 3.10
			• Secure data (STE/STU-III) (R)	• CJCSI 6215.01B
		VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
DSN Line Interfaces				
		Access	• Directory Number Identification (R)	• GSCR Section 2.1.1
			• Loop Start Line (R: 2-Wire Analog only)	• GSCR Section 5.2.1
			• Ground Start Line (R)	• GSCR Section 5.2.2
			• Alerting Signals and Tones (R)	• GSCR Section 5.5
			• WWNDP (R)	• GSCR Section 4.5
2-Wire Analog	Yes		• Origination Treatments (R)	• GSCR Section 4.1.1
			• Termination Treatment (R)	• GSCR Section 4.1.2
			• Release Treatment (R)	• GSCR Section 4.1.3
			• Interruption Treatment (R)	• GSCR Section 4.1.4
			• Connections (R)	• GSCR Section 4.1.5
			• Class of Service (C)	• GSCR Section 4.1.6
			• 2W user access (R: 2-Wire Analog only)	• GSCR Section 4.3.3
ISDN BRI NI 1/2 (ANSI T1.619a)	No		• Analog busy/idle (R: 2-Wire Analog only)	• GSCR Section 4.3.4.1
		Voice	• MOS (R)	• CJCSI 6215.01B
			• MLPP (R)	• GSCR Section 3 applicable paragraphs
			• Secure Calls (R)	• CJCSI 6215.01B
2-Wire Proprietary Digital	No	Facsimile	• Analog: TIA/EIA-465-A (R)	• DISR
		Data	• Modem (VBD) (R)	• CJCSI 6215.01B
			• 56 kbps switched data (C: BRI only)	• GSCR Section 3.10
			• 64 kbps switched data (C: BRI only)	• GSCR Section 3.10
			• NX56 synchronous BER (C: BRI only)	• GSCR Section 3.10
			• NX64 synchronous BER (C: BRI only)	• GSCR Section 3.10
			• Secure data (STE/STU-III) (R)	• CJCSI 6215.01B
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Common Features	No	• Selective call rejection (C) • Denied originating service (C) • Code restriction and diversion (C) • Call waiting (C) • Three-way calling (C) • Add-on transfer, conference calling, and call hold (C) • Call forwarding (C) • Call pick-up (C)	• GSCR Section 2.1.2 • GSCR Section 2.1.3 • GSCR Section 2.1.4 • GSCR Section 2.1.5 • GSCR Section 2.1.6 • GSCR Section 2.1.7 • GSCR Section 2.1.8 • GSCR Section 2.1.9
Attendant	No	• Initiate all precedence levels (C) • Visual display (C) • Override class of service (C) • Override busy line (C) • Call deflection (C) • Auto recall (C) • Waiting queue (C)	• GSCR Section 2.2.1 • GSCR Section 2.2.2 • GSCR Section 2.2.3 • GSCR Section 2.2.4 • GSCR Section 2.2.5 • GSCR Section 2.2.6 • GSCR Section 2.2.7
Public Safety	No	• Basic Emergency Service (911) (C) • Trace of terminating calls (C) • Outgoing call trace (C) • Tandem call trace (C) • Trace of a call in progress (C)	• GSCR Section 2.4.1 • GSCR Section 2.4.2 • GSCR Section 2.4.3 • GSCR Section 2.4.4 • GSCR Section 2.4.5
Preset Conferencing	No	• Support 10 bridges; 1 originator and 20 conferees per bridge (C) • Assign up to 20 address numbers per bridge (C) • Use KXX codes for bridge access (C) • Conference notification recorded announcement (C) • Auto retrial and alternate address (C) • Bridge release (C) • Lost connection (C) • Secondary conferencing (C) • Meet Me Conferencing (C) • Address translation (C)	• GSCR Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6.1 • GSCR Section 2.6.2 • GSCR Section 2.6.3 • GSCR Section 2.6.4 • GSCR Section 2.6.5 • GSCR Section 2.6.6 • GSCR Section 2.7
Nailed-up Connections	No	• Between any two like terminations (C) • PCM-24 and PCM-30, both CAS and CCS (C) • Supervision passed end-to-end for A/D or D/A (C) • Monitored and auto reconfigure (C) • Support at least 10% of circuits as nailed-up (C) • Non-preemptable (C)	• GSCR Section 2.8 • GSCR Section 2.8 • GSCR Section 2.8 • GSCR Section 2.8 • GSCR Section 2.8 • GSCR Section 2.8
PAT	No	• Classmark for/not for PAT screening (C) • 7 PAT mechanisms (C) • Outgoing call screening (C) • Functional structure (C) • Simultaneous calls limitation (C) • Overflow process (C) • Decrementing call-in-progress count (C) • Call treatment (C) • Queuing (C) • Attendant calls (C) • Operations measurement registers (C) • Maintenance and Administration of thresholds (C)	• GSCR Section 2.11.1 • GSCR Section 2.11.1 • GSCR Section 2.11.1.1 • GSCR Section 2.11.1.2 • GSCR Section 2.11.1.3 • GSCR Section 2.11.1.4 • GSCR Section 2.11.1.5 • GSCR Section 2.11.1.6 • GSCR Section 2.11.1.7 • GSCR Section 2.11.1.8 • GSCR Section 2.11.1.9 • GSCR Section 2.11.1.10
DSN Hotline Services	No	• Hotline restrictions (C) • Auto initiate (C) • Analog and digital (C) • Subscription basis (C) • Protected hotline calling (C)	• GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12.1-4

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Network Management	No	• Interfaces (C) • Measurements and data generation (C) • Fault management (C) • Configuration management (C) • Accounting management (C) • Performance management (C) • Network management controls (C) • Remote access (C)	• GSCR Section 9.1 • GSCR Section 9.2 • GSCR Section 9.3 • GSCR Section 9.4 • GSCR Section 9.5 • GSCR Section 9.6 • GSCR Section 9.7 • GSCR Section 9.8
ISDN Services	No	• Electronic Key Telephone Systems (EKTS) (C)	• GSCR Section 10, table 10-3
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R)	• GSCR Section 11.1.1.2 • GSCR Section 11.1.2.2
Reliability	Yes	• GR-512-CORE (R)	• GSCR Section 12.2
Security	Yes	• GR-815, STIGs, and DIACAP (replacement for DITSCAP) (R)	• GSCR Section 13
VoIP			
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • Class of Service (CoS) and Quality of Service (QoS) (R) • ITU-T G.711 PCM CODEC (R) • Traffic Engineering (R) • Security (R) • Network management (R) • Line timing (R) • Internal Clock (R) • Latency ≤ 60 milliseconds (R) • Packet Loss (R) • IPv6 capable (R)	• GSCR Appendix 3, para. 3.2.1 • GSCR Appendix 3, para. 3.3.2.1 • GSCR Appendix 3, para. 3.2.2 • GSCR Appendix 3, para. 3.3.4.4.2 • GSCR Appendix 3, para. 3.2.4 • GSCR Appendix 3, para. 3.3.4.2 • GSCR Appendix 3, para. 3.2.6 • GSCR Appendix 3, para. 3.2.6 • GSCR Appendix 3, para. 3.2.7 • GSCR Appendix 3, para. 3.3.1.3 • GSCR Appendix 3, para 3.2.8 and Section 1 para. 1.7

Table 3. PBX 1 Requirements (continued)

Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN[1]	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C)	• CJCSI 6215.01B • CJCSI 6215.01B • CJCSI 6215.01B
DRSN[2]	Yes	Access	• Alerting Signals and Tones (R) • Call Processing (R) • Call Treatments (R) • Analog busy/idle (R)	• GSCR Section 5.5 • GSCR Section 4.4 • GSCR Section 4.1 • GSCR Section 4.3.4.1
		Voice	• MOS (C) • MLPP (C) • Secure calls (C)	• CJCSI 6215.01B • GSCR Section 3 applicable paragraphs • CJCSI 6215.01B

LEGEND:

2W	- 2-Wire	FTR 1080B-2002	- Video Teleconferencing Services	PAT	- Precedence Access Threshold
A/D	- Analog to Digital Conversion	G.711	- Standard for PCM of Voice Frequencies	PBX 1	- Private Branch Exchange 1
ANSI	- American National Standards Institute	GR	- Generic Requirement	PCM	- Pulse Code Modulation
BER	- Bit Error Ratio	GR-512-CORE	- LSSGR: Reliability, Section 12	PCM-24	- Pulse Code Modulation - 24 Channels
BRI	- Basic Rate Interface	GR-815	- Generic Requirements For Network Element/Network System (NE/NS) Security	PCM-30	- Pulse Code Modulation - 30 Channels
C	- Conditional			PRI	- Primary Rate Interface
CAS	- Channel Associated Signaling	GSCR	- Generic Switching Center Requirements	PSTN	- Public Switched Telephone Network
CCS	- Common Channel Signaling	H.320	- Narrow-band visual telephone systems and terminal equipment	Q.955.3	- ISDN Signaling Standard for E1 MLPP
CJCSI	- Chairman of the Joint Chiefs of Staff Instruction			R	- Required
		IPv6	- Internet Protocol version 6	SS7	- Signaling System 7
CODEC	- Coder/Decoder	ISDN	- Integrated Services Digital Network	STE	- Secure Terminal Equipment
D/A	- Digital to Analog Conversion	IT	- Information Technology	STIGs	- Security Technical Implementation Guides
DIACAP	- DoD Information Assurance Certification and Accreditation Process	ITU-T	- International Telecommunication Union-Telecommunication Standardization Sector	STU-III	- Secure Telephone Unit -3rd generation
				T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DISR	- DoD IT Standards Registry	kbps	- kilobits per second		
DITSCAP	- DoD IT Security Certification and Accreditation Process	KXX	- K= any number 2-8; X= any number 1-9	T1.619a	- SS7 and ISDN MLPP Signaling Standard for T1
		LSSGR	- Local Access and Transport Area (LATA) Switching Systems Generic Requirements		
DoD	- Department of Defense			TIA	- Telecommunications Industry Association
DP	- Dial Pulse	Mbps	- Megabits per second	TIA/EIA-465-A	- Group 3 Facsimile Apparatus for Document Transmission
DRSN	- Defense Red Switch Network	MFR1	- Multi-Frequency Recommendation 1		
DSN	- Defense Switched Network	MLPP	- Multi-Level Precedence and Preemption	VBD	- Variable bit data
DTMF	- Dual Tone Multi-Frequency	MOS	- Mean Opinion Score	VoIP	- Voice over Internet Protocol
E1	- European Basic Multiplex Rate (2.048 Mbps)	NI 1/2	- National ISDN Standard 1 or 2	VTC	- Video Teleconferencing
EIA	- Electronic Industries Alliance	NX56	- Data format restricted to multiples of 56 kbps	WWNDP	- Worldwide Numbering and Dialing Plan
FTR	- Federal Telecommunications Recommendation	NX64	- Data format restricted to multiples of 64 kbps		
		para	- paragraph		

NOTES:
1 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
2 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi.

6. The JITC point of contact is Mr. Edward Mellon, DSN 879-5159, commercial (520) 538-5159, FAX DSN 879-4347, or e-mail to edward.mellon@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 0716901.

FOR THE COMMANDER:

for J. T. Schutte

Enclosure a/s

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ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, Memo, "Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.2(3) Service Release (SR) 3, with Internetwork Operating System (IOS) Software Release 12.4(9) T1," 4 March 2008

(d) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001

(d) Defense Information Systems Agency (DISA), "Defense Switched Network (DSN) Generic Switching Center Requirements (GSCR), Errata Change 2," 14 December 2006, Revised 27 March 2007

(e) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006